Sub B1

WHAT IS CLAIMED IS:

1. A method for reducing the amount of computations required to create a sound signal representing one or more sounds originating at a plurality of discrete positions in space, where the signal is to be perceived as simulating one or more sounds at one or more selected positions in space with respect to a listener, comprising the steps of:

(a) determining a characteristic function for a position in space at which sound is to be received, wherein said characteristic function represents a head-related impulse response;

(b) applying said characteristic function as a filter to the signal representing sound to produce a filtered signal; and

(c) converting the filtered signal to a sound wave and producing the sound wave for a listener.

2. The method of claim 1 wherein said characteristic function further comprises information concerning the environment in which sound is to be perceived.

3. The method of claim 1 wherein said characteristic function is a spatial feature extraction and regularization model.

4. The method of claim 3 wherein said spatial feature extraction and regularization model comprises a spatial component and a temporal component.

5. The method of claim 4 wherein said temporal component comprises a summed matrix of a predetermined number of eigen vectors.

6. The method of claim 5 wherein said predetermined number of eigen vectors is of a range from 3 to 16.

7. The method of claim 5 wherein said spatial and temporal components are determined via a Karhunen-Loeve Expansion.

Sub C1

8. The method of claim 1 wherein the spatial characteristic function is determined for a selected number of N samples and a selected number of M eigen values and wherein the model filter function for an azimuth position θ and an elevation position ϕ of sound origination in a spherical coordinate system about the position of sound measurement as the origin has the form

$$y(n) = \sum_{m=1}^{M}\left[\sum_{k=1}^{K} w_m(\theta_k, \varphi_k) s_k(n)\right] q_m(n). \quad 9(c)$$

Sub C1

where s represents a sound source, K represents the number of independent sound sources, $w_m(\theta,\varphi)$ are the weighing factors, and $q_m(n)$ is a vector representing an orthonormal basis for a head-related impulse function.

Sub B2

9. Apparatus for providing sound created by a sound source to a listener which simulates the sound source at a selected position in space with respect to the listener, comprising:

(a) an input for receiving a signal representing a sound;

(b) a left channel and a right channel, wherein each channel comprises a filter array for applying a filter to the signal received by the input to provide a filtered signal, the filter comprising a linear function which comprises a head-related impulse response;

(c) an output for converting the filtered signals from said channels to a binaural sound and for producing the sound for a listener.

10. The apparatus of claim 9 wherein said linear function comprises a spatial feature extraction and regularization model.

11. The apparatus of claim 9 wherein said linear function includes a spatial component, said spatial component comprising signal delay and attenuation for simulating reflected sound created by surfaces of a sound reproduction environment.

12. The apparatus of claim 9 wherein said linear function includes a temporal component, said temporal component comprising a summed array of a predetermined number of eigen filters.

13. The apparatus of claim 9 further comprising:

an environment input for receiving information concerning a listening environment to be simulated and relative position of a listener;

a calculator for receiving the information from said environment input, and calculating attenuation and time delays to simulate said environment and said listener position;

wherein the output of said calculator is input into said filter array as factors for said linear function.

14. The apparatus of claim 13 further comprising a summed array of a predetermined number of eigen filters attached to said signal input and receiving the signal therefrom, wherein said eigen filters introduce time delays into said signal.

15. The apparatus of claim 14 wherein said filter array comprises:

(i) a plurality of source placement arrays, wherein each source placement array receives the output of a single eigen filter and filters said signal in accordance with a spatial characteristic function and the output of said calculator;

(ii) a summer for summing the output of the source placement arrays; and

(iii) a timer and delay for receiving the summed output signal from said summer and a delay count from said calculator.

Sub B3

16. An apparatus for providing sounds created by a plurality of sound sources to a listener which simulates the origin of each sound at a selected position in space with respect to the listener, comprising:

(a) an environment input for receiving information concerning a listening environment to be simulated and relative position of a listener;

(b) a calculator for receiving the information from said environment input, and calculating attenuation and time delays to simulate said environment and said listener position;

(c) a signal input for receiving a signal representing a sound;

(d) a left channel and a right channel attached to said calculator and receiving said calculation of attenuation and time delay therefrom, and also attached to said signal input and receiving said sound signal from said signal input, each channel comprising:

(i) a source placement array for filtering said sound signal in accordance with a spatial characteristic

function, wherein said spatial characteristic function is a head-related impulse response;

(ii) an plurality of eigen filters attached to said source placement array and receiving the signal therefrom, wherein said eigen filters introduce time delays into said signal; and

(iii) a signal output for attaching a speaker to the apparatus, attached to said plurality of eigen filters for receiving and summing the signals therefrom.

17. The apparatus of claim 16 further comprising a plurality of signal inputs for receiving a plurality of signals representing a plurality of sounds, wherein each channel further comprises a plurality of source placement arrays, each of said source placement arrays mated to a single signal input, and a plurality of summers for receiving and summing the signal from each source placement array and for outputting the summed signal into said temporal filter.

18. The apparatus of claim 16 wherein said plurality of eigen filters is of a range from 3 to 16.

19. The apparatus of claim 16 further comprising a delay buffer for introducing a temporal delay into said signal, wherein said delay buffer receives the signal from said sound input and outputs the delayed signal into each channel.

20. The apparatus of claim 16 wherein said apparatus further comprises a cross-talk canceler for filtering cross-talk in said signal prior to reproduction by said speakers.

add c37